IN THE CLAIMS

Please amend the claims as follows:

Claim 1 (Currently Amended): A sampling rate converter comprising:

an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to [[an]] the output signal of the up sampler,

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the up sampler, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 2 (Previously Presented): A sampling rate converter as set forth in claim 1, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 3 (Previously Presented): A sampling rate converter as set forth in claim 1, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the low pass filter.

Claim 4 (Previously Presented): A sampling rate converter as set forth in claim 1, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 5 (Previously Presented): A sampling rate converter as set forth in claim 1, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 6 (Currently Amended): A sampling rate converter comprising:

an up sampler for <u>receiving an input signal at frequency Fsi and for</u> inserting U-1 zero points between sample signals and raising a sampling frequency U-fold <u>to output an output signal at frequency UFsi</u>,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to [[an]] <u>the</u> output signal of the up sampler,

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the up sampler, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes a filter coefficient, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 7 (Original): A sampling rate converter as set forth in claim 6, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

Claim 8 (Previously Presented): A sampling rate converter as set forth in claim 6, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 9 (Previously Presented): A sampling rate converter as set forth in claim 6, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 10 (Currently Amended): A sampling rate converter comprising:

an up sampler for <u>receiving an input signal at frequency Fsi and for</u> inserting U-1 zero points between sample signals and raising a sampling frequency U-fold <u>to output an output signal at frequency UFsi</u>,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to [[an]] the output signal of the up sampler,

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the up sampler, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 11 (Previously Presented): A sampling rate converter as set forth in claim 10, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 12 (Previously Presented): A sampling rate converter as set forth in claim 10, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the low pass filter.

Claim 13 (Previously Presented): A sampling rate converter as set forth in claim 10, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 14 (Previously Presented): A sampling rate converter as set forth in claim 10, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 15 (Currently Amended): A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing the convolution processing of input sample signals and the poly-phase filters decomposed to the poly-phases,

a plurality of up samplers for <u>receiving input signals at frequency Fsi and for</u> inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold <u>to output output signals at frequency UFsi</u>,

an adding means for generating a signal after adding all signals by adjusting a propagation time of <u>the</u> output signals of the plurality of up samplers,

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from the linear interpolation, and

a low pass filter providing either low pass filtered signals to the plurality of up samplers, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 16 (Previously Presented): A sampling rate converter as set forth in claim 15, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 17 (Previously Presented): A sampling rate converter as set forth in claim 15, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the low pass filter.

Claim 18 (Previously Presented): A sampling rate converter as set forth in claim 15, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 19 (Previously Presented): A sampling rate converter as set forth in claim 15, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 20 (Currently Amended): A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for receiving input signals at frequency Fsi and for inserting U-1 zero points between output signals of corresponding the convolution processing units and raising the sampling frequency U-fold to output output signals at frequency UFsi,

an adding means for generating a signal after adding all signals by adjusting a propagation time of the output signals of the plurality of up samplers,

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered signals to the plurality of up samplers, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and an impulse response becomes the filter coefficient, and

the filter coefficient is set by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 21 (Original): A sampling rate converter as set forth in claim 20, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

Claim 22 (Previously Presented): A sampling rate converter as set forth in claim 20, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 23 (Previously Presented): A sampling rate converter as set forth in claim 20, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 24 (Currently Amended): A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for receiving input signals at frequency Fsi and for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold to output output signals at frequency UFsi,

an adding means for generating a signal after adding all signals by adjusting a propagation time of the output signals of the plurality of up samplers,

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the plurality of up samplers, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 25 (Previously Presented): A sampling rate converter as set forth in claim 24, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 26 (Previously Presented): A sampling rate converter as set forth in claim 24, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the low pass filter.

Claim 27 (Previously Presented): A sampling rate converter as set forth in claim 24, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 28 (Previously Presented): A sampling rate converter as set forth in claim 24, wherein the low pass filter prevents an imaging component from occurring and a non-original

frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 29 (Currently Amended): A sampling rate converter comprising:

an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter,

a linear interpolation block for finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered signals to the convolution processing unit, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 30 (Previously Presented): A sampling rate converter as set forth in claim 29, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer

obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 31 (Previously Presented): A sampling rate converter as set forth in claim 29, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the low pass filter.

Claim 32 (Previously Presented): A sampling rate converter as set forth in claim 29, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 33 (Previously Presented): A sampling rate converter as set forth in claim 29, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 34 (Original): A sampling rate converter as set forth in claim 29, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

Claim 35 (Currently Amended): A sampling rate converter comprising:

an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter,

a linear interpolation block for finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered signals to the convolution processing unit, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 36 (Original): A sampling rate converter as set forth in claim 35, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 37 (Previously Presented): A sampling rate converter as set forth in claim 35, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 38 (Previously Presented): A sampling rate converter as set forth in claim 35, wherein the low pass filter prevents an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 39 (Original): A sampling rate converter as set forth in claim 35, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

Claim 40 (Currently Amended): A sampling rate converter comprising:

an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter,

a linear interpolation block for finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered signals to the convolution processing unit, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 41 (Previously Presented): A sampling rate converter as set forth in claim 40, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 42 (Previously Presented): A sampling rate converter as set forth in claim 40, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the low pass filter.

Claim 43 (Previously Presented): A sampling rate converter as set forth in claim 40, wherein the low pass filter prevents an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

Claim 44 (Previously Presented): A sampling rate converter as set forth in claim 40, wherein the low pass filter prevents an imaging component from occurring and a non-original

frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

Claim 45 (Original): A sampling rate converter as set forth in claim 40, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

Claim 46 (Currently Amended): A sampling rate conversion method comprising:

a first step of receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising the sampling frequency U-fold to output an output signal at frequency UFsi,

a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a low pass filter,

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, and

providing either low pass filtered sample signals through the low pass filter to the first step, or low pass filtering signals output of the third step through the low pass filter, wherein

the filter coefficient of the FIR filter is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 47 (Currently Amended): A sampling rate conversion method comprising:

a first step of receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising the sampling frequency U-fold to output an output signal at frequency UFsi,

a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length and an impulse response becomes the filter coefficient,

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the third step through the low pass filter, wherein

the filter coefficient of the FIR filter is calculated by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 48 (Currently Amended): A sampling rate conversion method comprising:

a first step of receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising the sampling frequency U-fold to output an output signal at frequency UFsi,

a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a low pass filter,

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, and

providing either low pass filtered sample signals to the first step through the low pass filter, or low pass filtering signals output of the third step through the low pass filter, wherein

the filter coefficient of the FIR filter is calculated by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the low pass filter.

Claim 49 (Currently Amended): A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of receiving an input signal at frequency Fsi and for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold to output an output signal at frequency UFsi,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals,

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the fourth step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 50 (Currently Amended): A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of <u>receiving an input signal at frequency Fsi and for</u> inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold <u>to output an output signal at frequency UFsi</u>,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals,

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the fourth step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 51 (Currently Amended): A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of <u>receiving an input signal at frequency Fsi and for</u> inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold <u>to output an output signal at frequency UFsi,</u>

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals,

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the fourth step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency points to be passed and a frequency response of the low pass filter.

Claim 52 (Currently Amended): A sampling rate conversion method comprising:

<u>receiving an input signal at frequency Fsi and inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,</u>

~~a first step of~~ selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter;

~~a second step of~~ performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the second step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency response of the low pass filter.

Claim 53 (Currently Amended): A sampling rate conversion method comprising:

<u>receiving an input signal at frequency Fsi and inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,</u>

~~a first step of~~ selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter;

~~a second step of~~ performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the second step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

Claim 54 (Currently Amended): A sampling rate conversion method comprising:

receiving an input signal at frequency Fsi and inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

~~a first step of~~ selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter;

~~a second step of~~ performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, and

providing either low pass filtered sample signals to the first step through a low pass filter, or low pass filtering signals output of the second step through the low pass filter, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency points to be passed and a frequency response of the low pass filter.

Claim 55 (Currently Amended): An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises:

an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to [[an]] the output signal of the up sampler,

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the up sampler, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the low pass filter.

Claim 56 (Currently Amended): An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for receiving input signals at frequency Fsi and for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold to output output signals at frequency UFsi,

an adding means for generating a signal after adding all signals by adjusting a propagation time of the output signals of the plurality of up samplers,

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, and

a low pass filter providing either low pass filtered sample signals to the plurality of up samplers, or low pass filtering signals output of the linear interpolation block, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing the weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the low pass filter.

Claim 57 (Currently Amended): An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises:

<u>an up sampler for receiving an input signal at frequency Fsi and for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold to output an output signal at frequency UFsi,</u>

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and

a low pass filter providing either low pass filtered signals to the convolution processing unit, or low pass filtering signals output of the selector, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of the low pass filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the low pass filter.